

A DUAL-CONVERSION MULTIMODE RECEIVE I.F./AF STRIP

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(PART 1)

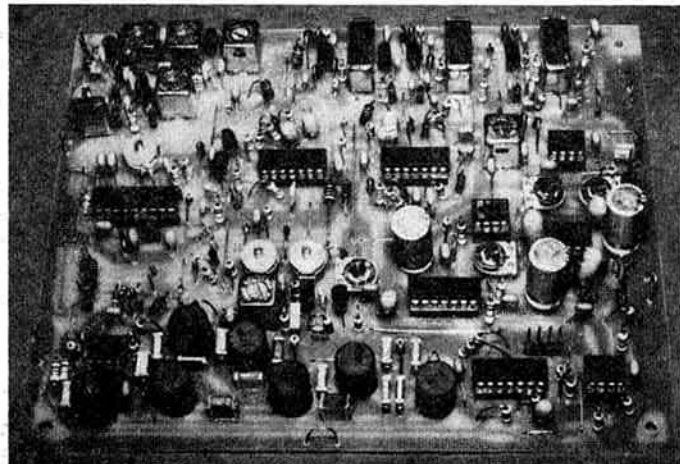
THERE SEEMS to be an absence in amateur literature of constructional articles on general-purpose receive i.f./af strips. For someone wanting to build an i.f. strip quickly and easily, perhaps because he wants to experiment with receiver front-ends, the choice is severely limited. The usual source for such information is to be found in articles on complete receivers or transceivers, but these usually reflect the interests of the designer as regards the modes of operation and the audio performance. Probably the nearest thing to a general purpose i.f. strip can be found in certain vhf receivers; three notable designs being described in [1], [2] and [3].

One i.f. strip which is available to amateurs in kit form is based on the G4CLF design [4] using Plessey SL1600 series ics. However, this cannot be described as general-purpose, as it was designed specifically for ssb and incorporates components for transmit as well as receive operation. As none of the published designs satisfied my requirements I decided to design a general purpose i.f./af strip, with the aims of producing a good-quality final design, and at the same time to improve performance and simplify design in certain areas—particularly in the use of dc switching and prewound inductors in the af filters.

Design criteria

Probably the most important choice that has to be made for any i.f. strip is the input frequency. For many years 9MHz has been a popular i.f. in amateur receivers. The exact reason for this choice is obscure, but it has the advantage (to an amateur) of allowing the two most popular amateur bands, 3.5 and 14MHz, to be received with a single local oscillator tuning 5.0-5.5MHz. Additionally, good image frequency rejection can be achieved on the hf bands without resort to complex pre-i.f. filters. High-quality crystal filters are available for 9MHz, but these have always been expensive. The cost of equipping a multimode i.f. strip with several 9MHz filters, each optimized for a particular mode, would be prohibitive to the average constructor.

More recently, 10.7MHz has become increasingly common as an i.f. for hf receivers. Initially this was encouraged by vhf receiver designers using surplus commercially-produced filters for nbfm. Filters suitable for ssb use are now available to amateurs (for example, the 10M22D, available from Cirkit Holdings) but these are still expensive when compared with 455kHz filters. The advent of digital frequency readout for receivers has also helped the move to 10.7MHz; modules are now available which can be programmed to give offsets of 10.7MHz, but not 9MHz.



Top view of the chassis

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The best value for money for i.f. filters can still be obtained at 455kHz. To combine the advantages of 10.7MHz from an image rejection point of view, and 455kHz because of the comparative cheapness of filters, a dual-conversion scheme was chosen. Also, 455kHz has the advantage that carrier insertion oscillators can be built with good stability without the need for crystals.

A complete list of the design criteria to be met is given below:

- (a) low-level i.f. input at 10.7MHz;
- (b) conversion to 455kHz for i.f. filtering;
- (c) selectable i.f. filters to suit modulation mode being received;
- (d) ssb, cw, a.m. and nbfm detectors;
- (e) fast attack, slow decay audio-derived agc;
- (f) agc line-driven S-meter;
- (g) variable level nbfm squelch;
- (h) selectable af filters;
- (i) all controls (including volume) dc controlled;
- (j) single-rail operation from nominally 12V;
- (k) all inductors pre-wound;
- (l) easily reproducible, therefore construction on single-sided pcb;
- (m) all components readily obtainable;
- (n) all connections to pcb via connectors, making removal of pcb easy;
- (o) sub-equipping permissible to suit particular preferences;
- (p) 9MHz permissible as first i.f. if desired.

All the above criteria have been met by the design described here.

The last two criteria are considered in more detail in the final chapter of this article.

Circuit description

A block diagram of the i.f./af strip is shown in Fig 1, and the complete circuit diagram is shown in Fig 2.

Input amplifier/converter

TR1 forms an input buffer stage with an input impedance equal to the value of resistor R1 (100k Ω). The drain of TR1 feeds a 10.7MHz filter consisting of two 10.7MHz i.f. transformers (T1, T2) connected back to back. TR2 is the 10.7MHz-455kHz converter, with the conversion oscillator output applied to the source and the 455kHz output selected by the 455kHz i.f. transformer T3.

The difference frequency between the two i.f.s (10,245kHz) is generated by TR3 and coupled to TR2 source via T4. Crystal X1 is the oscillator frequency determining element, and a degree of trimming can be achieved by adjusting trimmer C95. If the alternative conversion frequency of 11,145kHz is required, X1 should be replaced by a crystal of this frequency and T4 tuned for maximum output at 11,145kHz.

The gain of the input amplifier and converter, when loaded with the filter driver stages (TR4, TR6 and TR7), from 10.7MHz input to 455kHz output is approximately 24dB.

455kHz filter stages/output selection

The secondary of T3 couples the 455kHz output from the converter into the filter driver stages, one of which is provided for each filter F1-F3. No switching is incorporated on the input side of the filters, so that all three filters are driven at all times. A removable link, LK1, provides a useful access point for injecting signals into the filter drivers during testing (see later).

F1, a Murata CFM455F or NTKK LFC12, is used in the fm signal path. It has a -6dB bandwidth of 12kHz and a -50dB bandwidth of 24kHz. This filter (as do the two other ceramic i.f. filters) provides good performance at reasonable cost. The correct drive impedance for F1 (1.5k Ω) is provided by R12, the collector resistor of TR4. R13, the emitter resistor of TR4, is bypassed for ac by R14 and C13, which give an ac gain for TR4 of approximately 3. R15 matches the output impedance of F1

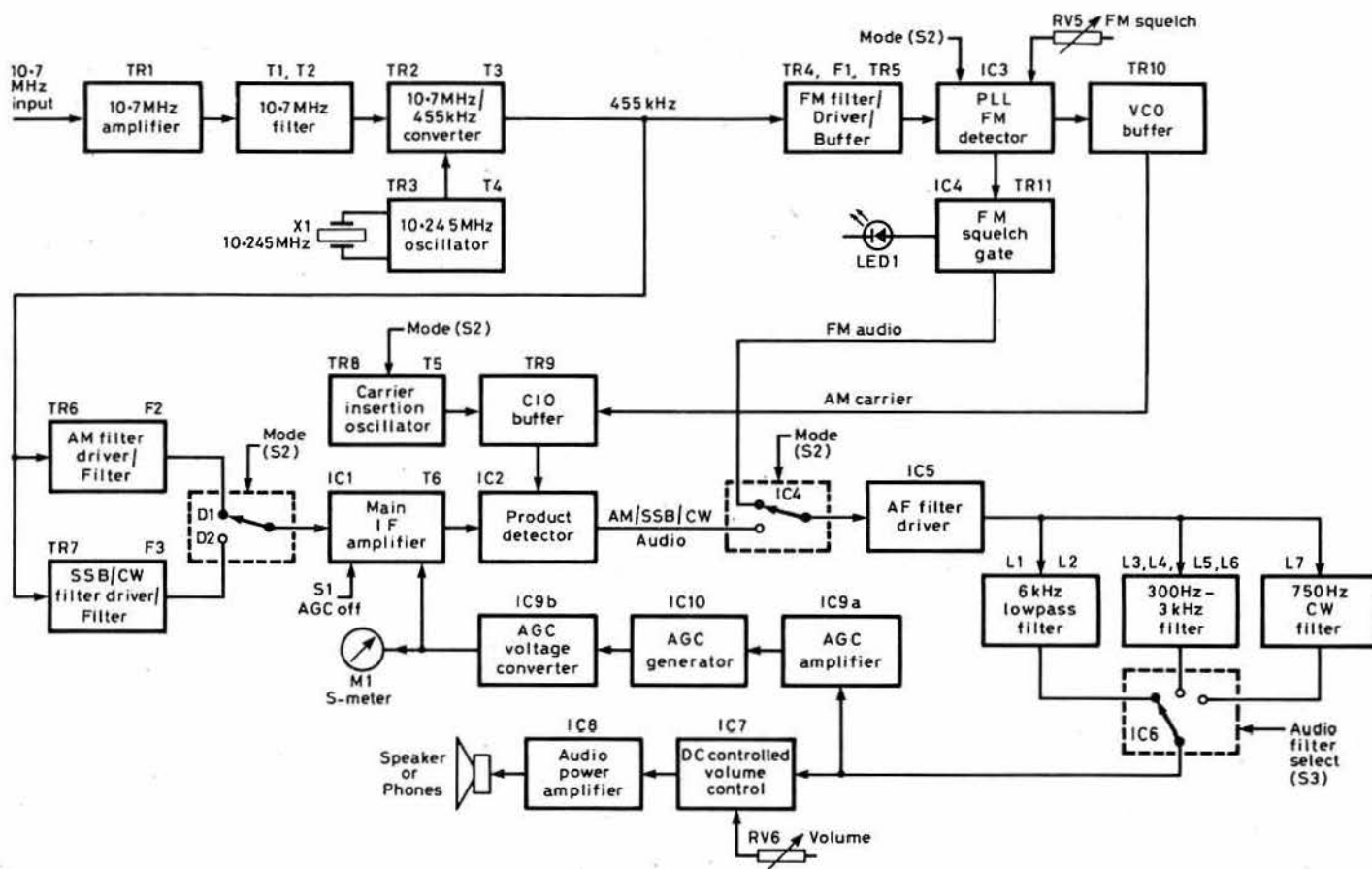


Fig 1. Block diagram

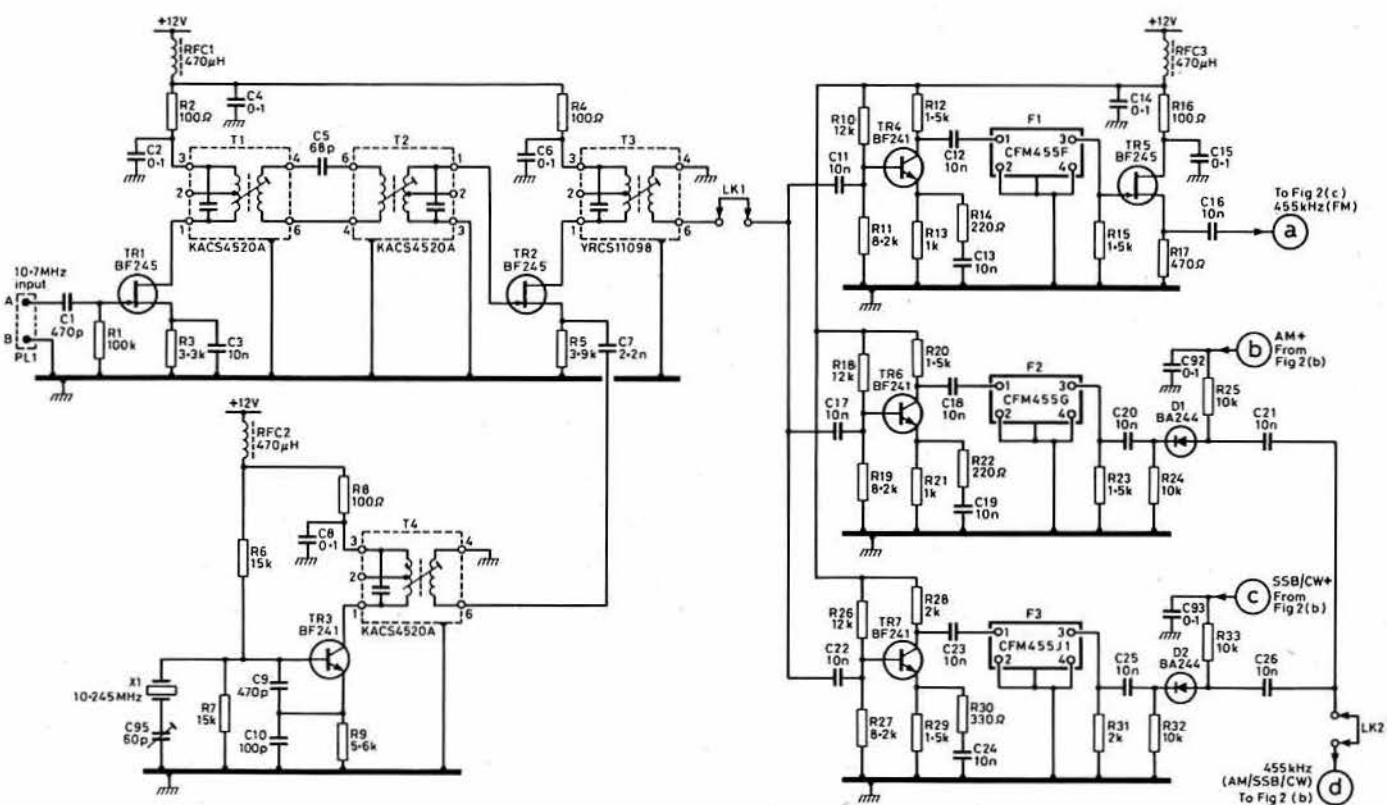


Fig 2. (a) Circuit diagram: 10-7MHz amplifier/filter, 10-7MHz-455kHz converter, 455kHz filters, drivers, switching

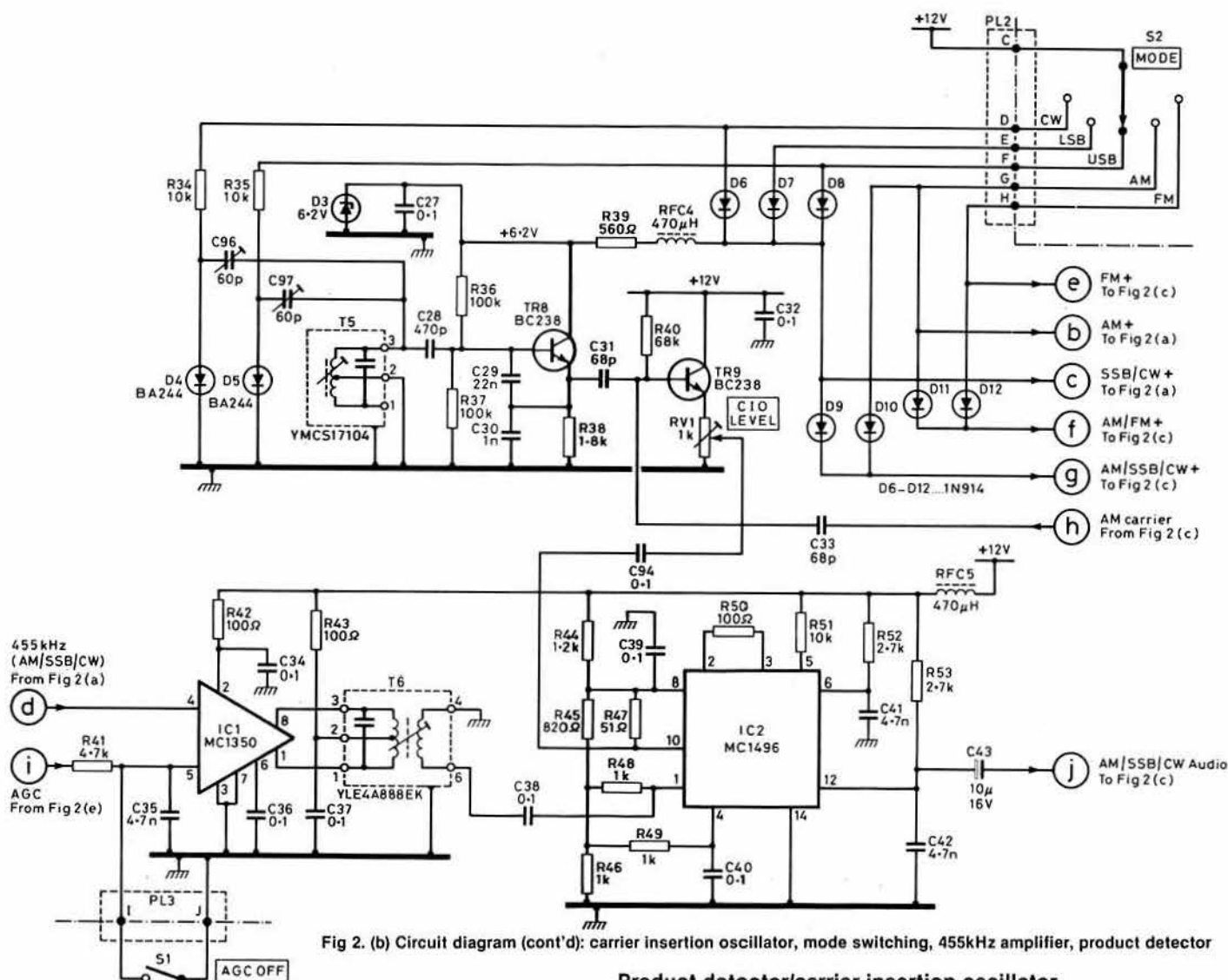


Fig 2. (b) Circuit diagram (cont'd): carrier insertion oscillator, mode switching, 455kHz amplifier, product detector

(again 1.5 k Ω) and also biases the gate of the fet source follower TR5. This stage buffers the correctly-matched output of F1 to the low impedance input of the fm detector IC3.

F2 is the a.m. filter. Here a Murata CFM4555G or NTKK LFC8 is used. This filter has a -6dB bandwidth of 8kHz and a -50dB bandwidth of 16kHz. R20 sets the drive impedance to the filter (again 1.5k Ω) and R23 terminates the filter correctly.

The ssb/cw filter is F3, a Murata CFM455J1. This filter is more expensive than F1 and F2 and has a -6dB bandwidth of 2.6kHz and a -50dB bandwidth of 8kHz. The input and output impedance of this filter is 2k Ω , matched by R28 and R31 respectively. Again the ac gain of the driver stage, TR7, is set at 3 by R30 and C24.

Selection of either the a.m. or ssb/cw filter output is determined by the MODE switch, S2. When in the a.m. mode, a positive voltage is applied to D1 (a BA244 switching diode) via R25, causing D1 to conduct through R24 and enabling the output of F2 into the input of the main i.f. amplifier, IC1. When in the usb, lsb or cw mode, D2 is made to conduct through R32 and R33, and the output of F3 is selected. Link LK2 gives another useful point at which test signals can be injected. No buffering is required between the outputs of F2 and F3 and the input of IC1, as the input impedance of IC1 is sufficiently high not to significantly affect the termination impedance seen by the filters.

Main i.f. amplifier

IC1, the main i.f. amplifier, is a Motorola MC1350. This ic has a maximum gain of 45dB and an agc range of 65dB [5]. AGC is applied to pin 5 and can be disabled, holding IC1 at full gain, by earthing pin 5 via switch S1. The push-pull output of IC1 is taken via T6 to the product detector, IC2. Current is supplied to the output transistors of IC1 via the centre tapped primary of T6.

Product detector/carrier insertion oscillator

The product detector is an MC1496, biased for single-rail operation as described in the Motorola circuit data [6]. IC2 is used as a product detector in both the ssb/cw and a.m. modes. Its operation as an ssb/cw detector is well documented, but I have not previously seen this ic used as an a.m. detector in amateur literature.

The principle of operation as an a.m. detector is to inject a carrier into pin 10 of IC2 at exactly the signal carrier frequency. This can be achieved in this application because the fm detector, IC3, contains a phase-locked loop which locks to exactly the incoming signal carrier frequency. Buffering is provided between the vco output of IC3 and pin 10 of IC2 by a fet buffer stage TR10 and emitter follower TR9, which is also used to buffer the ssb/cw cio output when in the usb, lsb or cw mode.

In the a.m. mode, the vco frequency of IC3 produces a zero beat with the received a.m. carrier, and equal frequency beats with the two a.m. sidebands, at the output of IC2. This rather novel method of a.m. detection was chosen for two reasons: first, no separate a.m. detector has to be incorporated; and second, no switching between a.m. and ssb/cw detectors is required as in either mode the audio output appears on pin 12 of IC2.

When in the usb, lsb or cw mode, the carrier source is the ssb/cw carrier insertion oscillator formed by TR8 and associated components. Power is only applied to the ssb/cw cio when in the usb, lsb or cw mode, a stabilized supply being derived by D3 and R39. In the lsb mode the frequency of the cio is determined by the inductance and internal capacitor of T5. T5 is adjusted to set the lsb mode frequency approximately 1.5kHz above the i.f.

When in the cw or usb mode, D4 and D5 respectively conduct, placing trimmers C96 or C97 in parallel with T5. These lower the frequency of the cio. In the cw mode the cio can be set to either side of the i.f., but when in the usb mode the cio must be on the lower side of the i.f. The labelling of the modes as lsb or usb here is arbitrary, as the switch setting suitable to demodulate the incoming sideband depends on whether sideband reversal

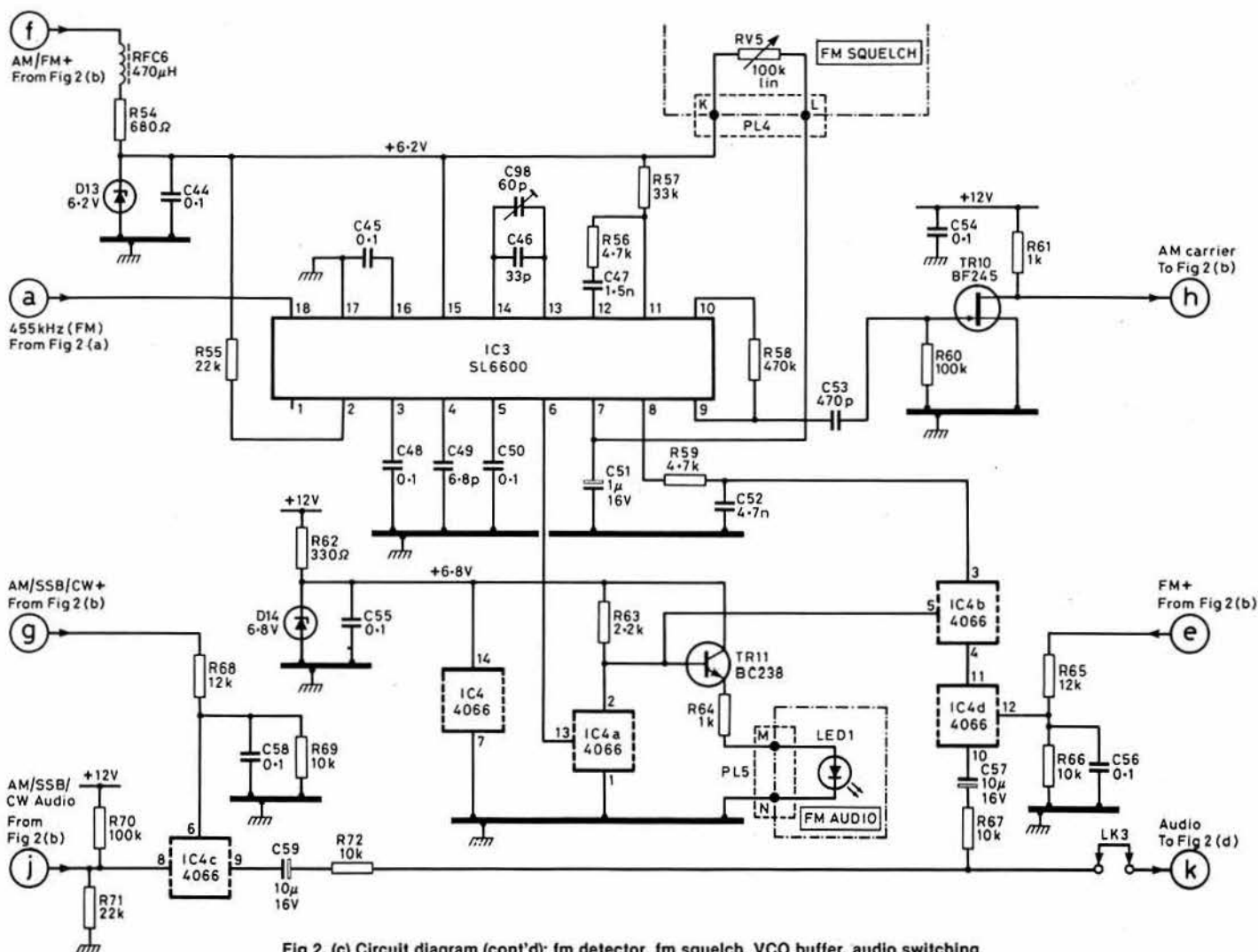


Fig 2. (c) Circuit diagram (cont'd): fm detector, fm squelch, VCO buffer, audio switching

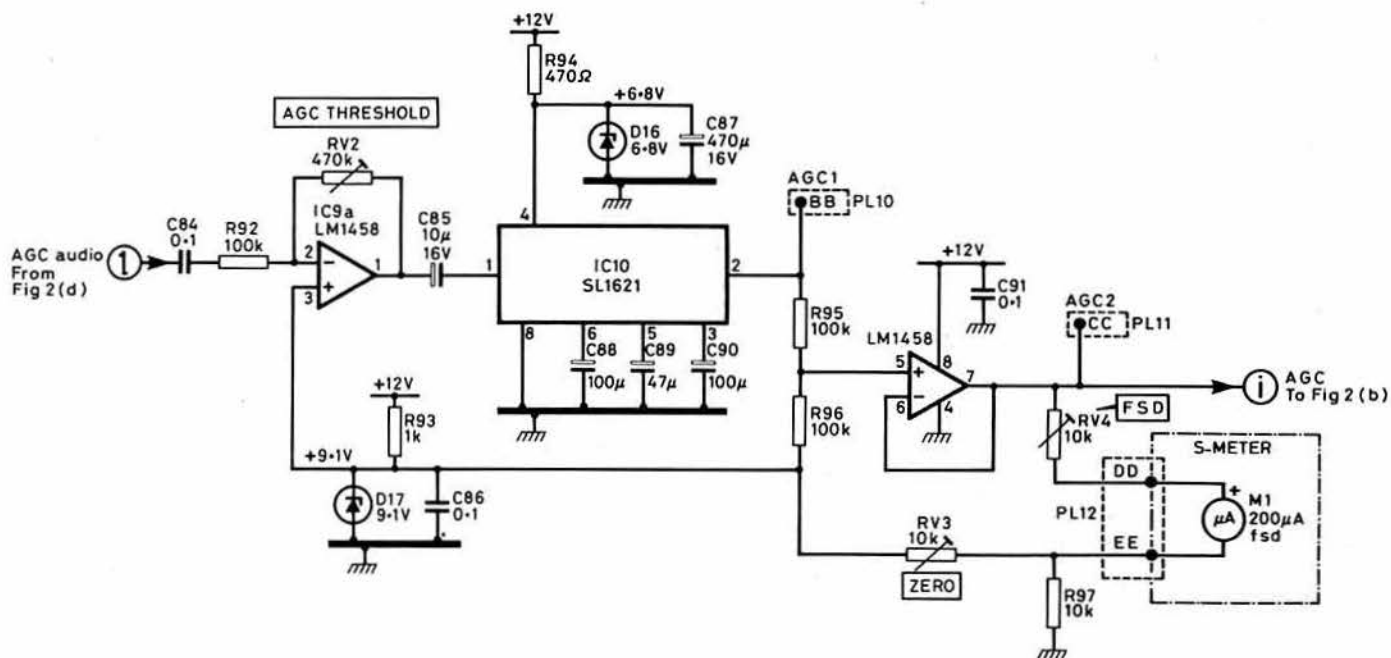


Fig 2. (e) Circuit diagram (cont'd): agc amplifier, agc controller, agc voltage converter

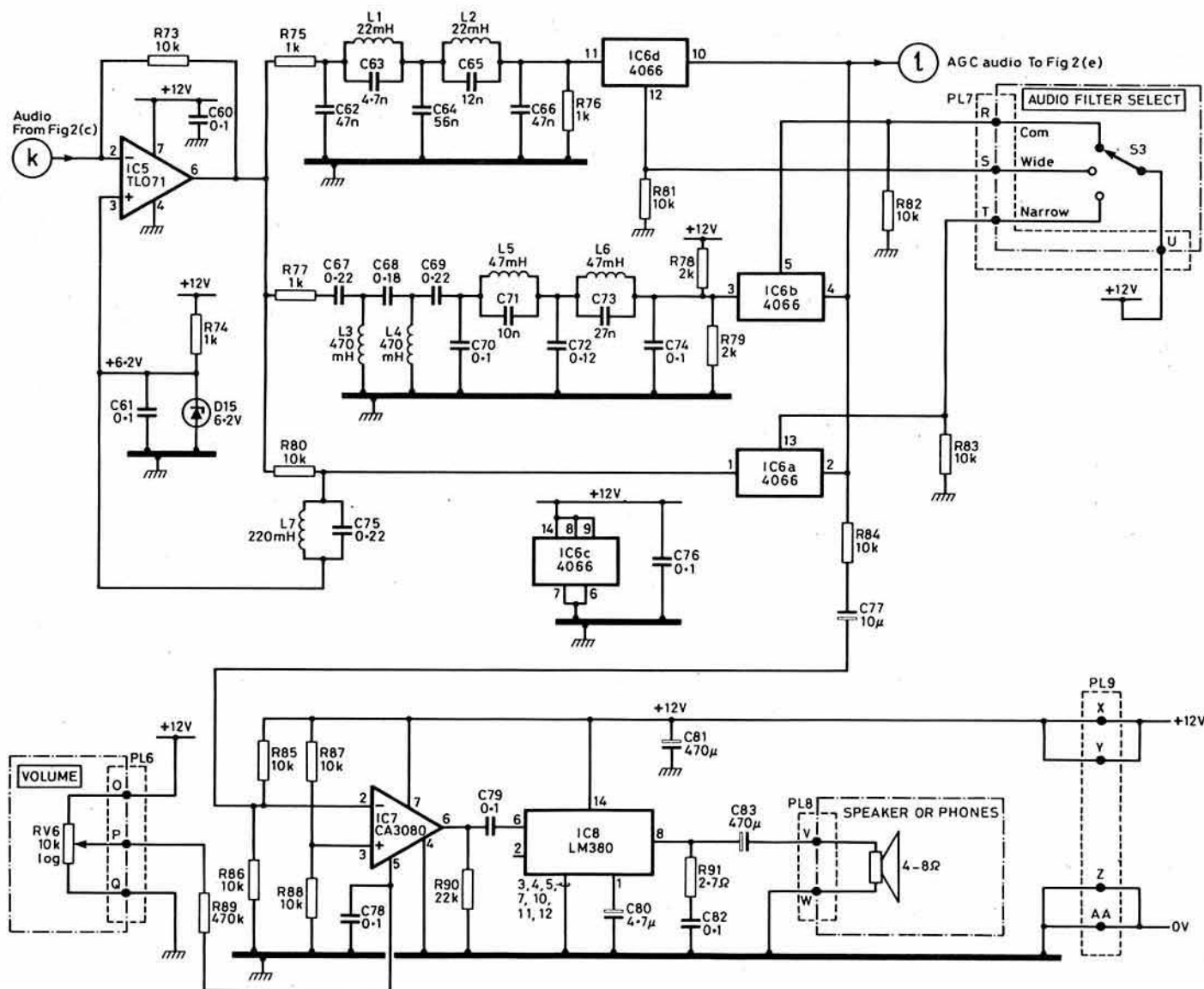


Fig. 2 (d) Circuit diagram (cont'd): audio filter driver, audio filters, audio filter switching, volume control, power amplifier

has taken place earlier in the i.f. strip or the receiver in which it is used due to the conversion frequencies used.

FM demodulator/squelch circuit

FM demodulation is provided by IC3, a Plessey SL6600 (or SL6601) low power (2.5mA maximum supply current) pll ic. In this application it is used as an amplifier/limiter/demodulator at 455kHz. This versatile ic can also be used as a single-conversion circuit from, say, 10.7MHz to 100kHz [7]. Power is applied to IC3 via D11 or D12 when S2 is in the a.m. or fm mode, and R54 and D13 generate a stabilised 6.2V supply. Adjustment of the free-running vco frequency is achieved by trimmer C98, and the pll filter components are C47 and R56. The demodulated fm audio output appears at IC3 pin 8 and passes via a simple lowpass filter formed by R59 and C52 to the input of the cmos analogue switch IC4b. IC4b is used as a squelch switch to mute the fm audio output when no fm signal is being received.

Pin 6 of IC3 is a squelch output whose voltage is almost at supply potential when the fm audio needs to be muted. Unfortunately this is the opposite polarity required by the control input of IC4b (pin 5). The squelch control signal is therefore inverted by IC4a which also switches TR11 to give a visual indication of the muted status via LED1. When the fm audio path is enabled, LED1 is illuminated and when it is disabled by squelch action, LED1 is extinguished.

The squelch level is adjusted by means of potentiometer RV5, connected between pin 7 of IC3 and its 6.2V supply rail. C51, between IC3 pin 7 and ground, determines the squelch attack time. Experimenters may wish to vary C51 (the recommended range of C51 is between 10nF and 10µF) to suit their preferences.

Detector output selection

Selection of either fm or a.m./ssb/cw audio is achieved by enabling IC4d or IC4c respectively. IC4d is enabled by placing a positive potential on pin 12, its control input, equal to almost its supply voltage, via the potential divider formed by R65 and R66. Alternatively, IC4c is enabled via diodes D6 and D9 in the cw mode, D7 and D9 in the lsb mode, D8 and D9 in the usb mode or D10 in the a.m. mode. Again IC4c is not enabled directly, but via a potential divider formed by R68 and R69.

Power is permanently supplied to IC4, derived from the 12V rail by R62 and the 6.8V zener D14.

It is important when using cmos analogue switches that the potential on an input never exceeds the supply voltage, otherwise permanent damage may occur. This is why IC4 is powered from a 6.8V supply (slightly greater than the supply of IC3) and the control inputs to IC4c and IC4d are potted down by R68/R69 and R65/R66. Another requirement of cmos analogue switches when used with a single supply rail is that the input waveform should not fall below 0V. If this does happen, breakthrough to the output of off switches can occur. The combination of R70 and R71 ensures that the input of IC4c is biased at approximately half-rail. A dc offset is present on the audio output of IC3 which provides bias to the inputs of IC4b and IC4d.

The audio outputs of IC4c and IC4d are merged via dc blocking capacitors C59 and C57 and R72 and R67, which set the gain for each audio path in conjunction with R73.

Again, a link is provided in the audio path for test access.

TO BE CONCLUDED

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(PART 2)

Audio filter driver/audio filters

The low-level audio at the junction of R67 and R72 is connected to the input of the low noise bifet operational amplifier IC5. The inputs and hence the output of IC5 are biased at approximately mid-rail by R74 and D15. IC5 has a gain of unity, and its main function is to act as a low output impedance driver stage to the three audio filters.

L1, L2 and associated capacitors form a fifth-order elliptic lowpass filter with a cut-off frequency of approximately 6.6kHz and a stopband attenuation of more than 45dB at frequencies greater than 9.7kHz. This filter has input and output impedances of 1k Ω which are matched by R75 and R76. In keeping with the design aim of using pre-wound components for all inductors, L1 and L2 are Toko type 10RB miniature inductors. Despite their small size and comparatively low Q at audio frequencies, these inductors have been found to be suitable for audio filter applications [8]. This filter is intended for use when listening to broadcast stations where high-frequency background noise is not excessive. A plot of the response of this filter is shown in Fig 3.

A communications-quality bandpass audio filter is formed by inductors L3-L6 and associated capacitors. This filter consists of a fifth-order highpass filter followed by a fifth-order lowpass filter. The total effect is a bandpass filter with cut-off frequencies of 450Hz and 3kHz, and a stopband attenuation of 45dB at frequencies less than 150Hz and greater than 4.4kHz. Again the input and output impedances are 1k Ω . The correct drive impedance is provided by R77, and the terminating impedance by R78 and R79 which appear to ac signals to be connected in parallel. This filter gives excellent results with weak ssb signals, cutting down the level of hum and hiss and eliminating shifted-frequency ssb signals which have not been eliminated by the i.f. filter. A plot of the response of this filter is shown in Fig 4.

L7 and C75 form a cw peaking filter centred at approximately 750Hz. This filter has a -3dB bandwidth of 200Hz and -20dB bandwidth of 2kHz, giving useful peaking of cw signals. A plot of the response of this filter is shown in Fig 5.

These Toko inductors are sensitive to dc currents, which can cause core saturation and hence reduction in inductance value. In the case of L1 and L2, the current is limited to approximately 3mA by R75 and R76. L3, L4, L5 and L6 have no direct connection to dc and so have no standing dc current. L7 is particularly sensitive to dc, so both ends have been biased to approximately 6.2V to prevent any current from flowing through it.

Audio filter output selection

The outputs of the audio filters are selected by IC6, another cmos analogue switch, controlled by switch S3. IC6 is permanently powered from the 12V rail. S3 supplies 12V to the control input of whichever analogue gate is selected. The control inputs to the unselected gates are pulled down to 0V by R81, R82 or R83.

DC bias for the signal input to IC6d, pin 11, is provided by the

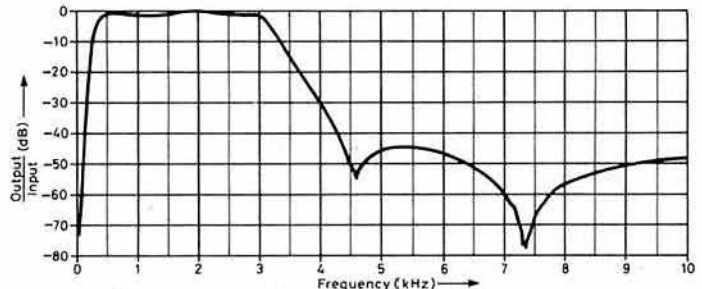


Fig 4. Communications bandwidth audio filter response

potentiometer effect of R75 and R76. Bias for IC6b pin 3 is provided by R78 and R79, this being the reason why a single 1k Ω resistor was not used to terminate the communications filter. IC6a pin 1 is biased to approximately 6.2V by R80 and L7.

The outputs of IC6a, IC6b and IC6d are commoned together and connected to the dc-controlled volume control circuit and the age generator.

Volume control/audio power amplifier

IC7 is an operational transconductance amplifier, manufactured by RCA. The output of this amplifier acts as a voltage-controlled current source, and the maximum gain is given by the product of the output transconductance (in this case typically 9.6mmho) and the load resistor, R90 [9]. R90 is necessary to convert the output current to a voltage to drive the audio power amplifier. With the value of R90 as in this application, the maximum voltage gain from IC7 is approximately 2.

Pin 5 of IC7 controls the amplifier bias current, enabling the output transconductance to be varied from zero to its maximum value. It is this feature which allows the device to be used as a volume control, the voltage gain being adjustable in this application from 0 to approximately 2 by potentiometer RV6.

Audio power amplification is provided by IC8, an LM380. The Zobel network, R91 and C82, on the output of IC8, prevents the possibility of rf oscillation under certain load conditions. Capacitor C80 bypasses an internal bias point within IC8.

In this application, with a supply of 12V, the maximum audio output is 1W rms into an 8 Ω load.

AGC circuit/S-meter

The output of whichever audio filter has been selected by S3 is sampled by C84 and R92 and amplified by IC9a, one half of an LM1458. Preset potentiometer RV2 allows the gain of IC9a, and hence the age threshold point, to be varied. Since IC9a is working from a single rail supply, its non-inverting input, pin 3, must be biased at some voltage above 0V for correct operation. A 9.1V supply is derived from the 12V rail by R93 and D17 for use later in the age circuit, so this is also used to bias IC9a.

IC10, a Plessey SL1621, generates the age timing. A 6.8V supply is provided by R94 and D16. The SL1621 can draw heavy transients of supply current, so a large value of capacitance, C87, is positioned on the supply rail to IC10. This ic was designed to give a fast-attack, slow-decay age characteristic, ideal for ssb reception. It also gives good results when

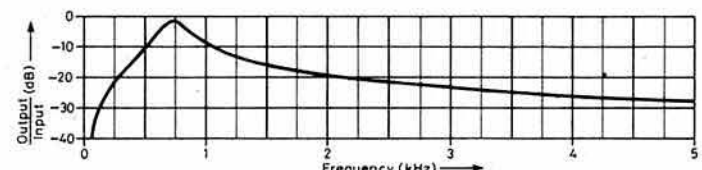


Fig 5. Narrow bandwidth audio filter response

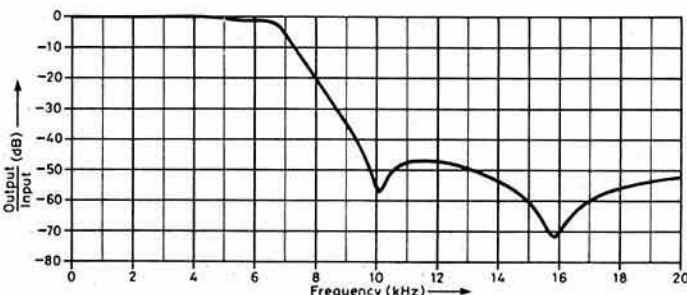


Fig 3. Wide bandwidth audio filter response

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receiving a.m. and cw. Full details of the operation of this ic can be obtained from the Plessey literature [10]. C88, C89 and C90 determine the agc timing characteristics.

Audio derived agc for a.m. works best when the modulation is fairly full and only brief pauses in the modulation occur. The a.m. mode included here was intended for broadcast use where both these criteria are generally met.

The SL1621 was designed for operation with the SL1600 series of rf/i.f. amplifiers and has an output agc voltage range of typically 1-5V. The MC1350, however, requires an agc voltage range of typically 5-7V; therefore, a voltage conversion has to be provided. This conversion is provided by R95 and R96. Assuming that the voltage at the D17 end of R96 is 9V, when IC10 pin 2 is at 1V, the junction of R95 and R96 will be at 5V. When IC10 pin 2 is at 5V, the R95/R96 junction will be at 7V. For voltages on IC10 pin 2 between 1 and 5V, the R95/R96 junction will track linearly between 5 and 7V, which is the required agc range. IC10 is only specified as sourcing current from pin 2 and, since in this application it is sinking current, the values of R95 and R96 have been made quite high to reduce the current into IC10 pin 2 to a minimum.

To boost the current available from the agc line to the MC1350, IC9b is configured as a non-inverting buffer whose output follows the voltage of R95/R96. The points labelled AGC1 and AGC2 give access for control of the front-end amplifier(s) of the receiver in which the i.f. strip is installed. This assumes that the voltage range of AGC1 or AGC2 is compatible to the front-end devices: if neither is suitable, an external voltage conversion will be necessary.

Drive for an S-meter is derived from the output of IC9b, pin 7. Preset potentiometers RV3 and RV4 allow the zero and full scale deflection of the meter to be set. The values of RV3, RV4 and R97 are suitable for a meter with a 200 μ A movement.

Supply decoupling

In the preceding description, no mention has been made of the supply decoupling arrangements. To prevent rf from being transferred among the circuits on the supply rails, use is made of rf chokes, RFC1-RFC6, and 0.1 μ F decoupling capacitors. These capacitors are also used to decouple the zener diodes to eliminate noise generated by these devices.

Construction

All the components for the i.f./af strip, apart from the external controls S1-S3, RV5, RV6, LED1 and M1, are mounted on a single-sided pcb. Fig 6 shows the pcb tracking, Fig 7 the drilling plan, and Fig 8 the component positions.

It is always difficult to describe the construction of a piece of electronic equipment. The author runs the risk of either insulting the intelligence of the experienced constructor or leaving the relative beginner with insufficient information. If the reader feels that more information is required, for example on the production of the pcb or the tools required, previous issues of *Radio Communication* or other magazines should be consulted. A particularly comprehensive description of the construction of a similar pcb was given for the RX80 Mk2 receiver [11].

Drill all the holes on the pcb to 0.85mm and then open out the fixing holes for the board to 3mm. Open out the holes for the can lugs on the transformers and filters, the connectors, the preset potentiometers and the trimmer capacitors. It is worthwhile to check carefully that all holes have been drilled to the correct size before mounting any components, otherwise components can be damaged if some holes have been missed and have to be drilled later.

In my opinion, it is best to start inserting and soldering components at the top left-hand corner of the pcb (when viewed from the component side)

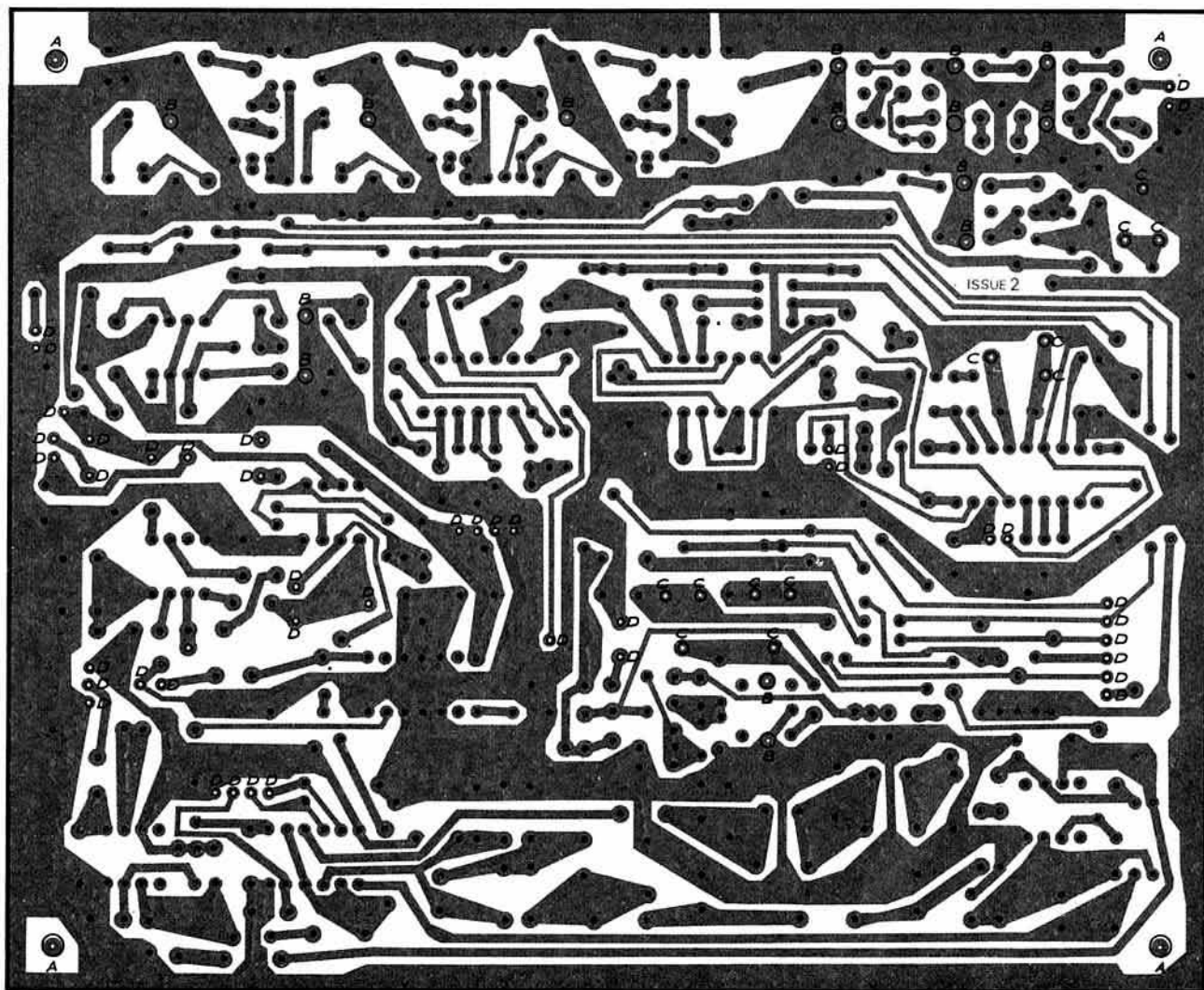


Fig 6. PCB layout

and work down to the bottom right-hand corner. Mount the components as they occur on the placement diagram, rather than mounting, say, the resistors first, then the capacitors, and so on. I have found that this method results in less mistakes than mounting one component type at a time. Use an earthed soldering iron and take the usual handling precautions with the fets.

To prevent the placement diagram (Fig 8) from becoming too cluttered, all resistors have been shown standing up; capacitors and diodes have been represented diagrammatically so that, where relevant, their polarities can be indicated. The best position for each component should be found, lying down on the board where possible.

Sockets were used for all the ics on the prototype, and no instability problems were encountered later. This contrasts with the usual advice given for this type of circuit, but it helps greatly when testing if a suspect ic can be changed easily.

For coupling and decoupling, $0.1\mu\text{F}$ capacitors are used extensively, and in most positions a physically fairly small component is required. The prototype used the Mullard 50V X7R component (available from Farnell). Another component which seems to be suitable is the Siemens type B37449, available from Electrovalue. Siemens polyester capacitors are ideal for use in the audio filter area, being very compact and having a lead spacing of 7.5mm . Tantalum beads were used for the other capacitors, apart from the three $470\mu\text{F}$ electrolytics. Here radial capacitors with 5mm lead pitch are required.

There is a total of 27 links on the board, which should be inserted at the same time as the other components—apart from links LK1, LK2 and LK3, which should be omitted at the construction stage and added later.

On the prototype board, connections were made to the external controls via 0.1in pitch connectors which are available from Cirkit Holdings. Alternatively, wires can be soldered directly into the pcb. The external connections are shown in Fig 10.

If wire is available with several colours of insulation, do not use the same colour twice on the same connector. For example, six colours would be required to wire all the connections to PL2 in different colours. This technique helps greatly in reducing the possibility of wiring errors. S1 and

LED1 are optional, so if they are not fitted simply omit the connectors. Similarly, connections to PL10 and PL11 (AGC1 and AGC2) may not be required. The leads to the speaker (or phones) are best done as a twisted pair of wires.

Cable shells and crimp terminals (again available from Cirkit Holdings) are used to connect the wires from the external controls to the on-board connectors. Do not rely solely on a crimped connection to the crimp terminals, but pinch the wire onto the terminal with pliers and then solder it. To ensure a good power supply connection, two connectors for each polarity are provided and should be commoned together.

It goes without saying that the board should be checked very carefully after completion, particularly the polarity of the diodes and capacitors. The board layout has been arranged that all the ics are orientated the same way round, and this helps to avoid mistakes. Similarly, check the external connections to the board, particularly the power supply polarity. If the board is being used with an external power supply, it does no harm to connect a diode in series with the supply leads and this can prevent wiring mistakes from being expensive.

Because all the controls to the board are simply dc levels, the lengths of leads to the front-panel controls are not critical. The board can be positioned wherever it is convenient. Connection to the i.f. input should be made with miniature coaxial cable with the outer conductor earthed at both ends.

Testing and alignment

The exact method of testing and alignment of this unit depends on what test equipment is available and whether it is tested on its own or when installed in a receiver. Experienced constructors may find it possible to complete alignment with the use of no test equipment at all, relying on a receiver front-end to generate an input signal at 10.7MHz , and their ears to detect when the transformers are peaked and the oscillators are at the correct frequency. For the sake of this description it will be assumed that a method of frequency measurement is available (at around 10MHz and 455kHz). This may be a digital frequency meter or a general coverage receiver. Also,

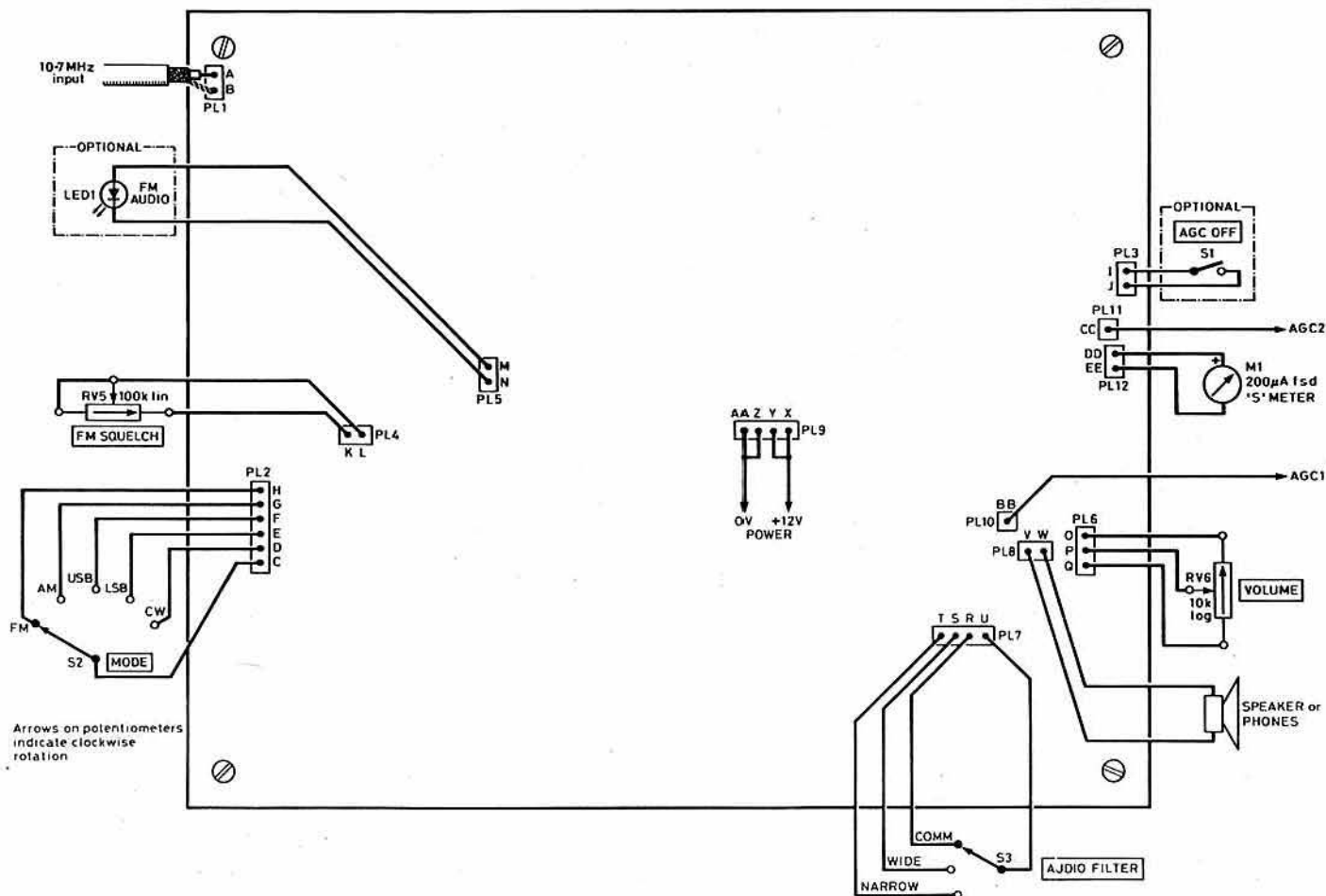


Fig 10. Wiring diagram of external connections

a signal generator is assumed to be available, working at audio frequencies, 455kHz and 10.7MHz. An oscilloscope and high input impedance voltmeter are assumed to be available for making measurements, though in many cases the ear will give a similar indication that a correct setting has been obtained.

If available, connect a milliammeter (200mA minimum) in series with the +12V lead. Set all the preset potentiometers (RV1-RV4) to mid-way. Set the volume control (RV6) and the fm squelch control (RV5) fully anticlockwise. Set S3 to communications filter.

Now follow this procedure:

(1) Apply the +12V supply. In usb, lsb or cw mode, the supply current should be approximately 130mA. In a.m. or fm mode, the current should be approximately 125mA. If the current is greatly different from these figures, switch off and check for a wrong connection or short circuit on the pcb.

(2) Now set the oscillators to the correct frequencies. Monitor T4 pin 6 and adjust C95 for a frequency of 10.245kHz. Rotate the core of T4 (using the correct trimmer tool) for maximum output. Approximately 1V peak-to-peak should be obtained at T4 pin 6.

Monitor IC2 pin 10, switch to lsb mode and rotate the core of T5 for a frequency of 456.5kHz.

Switch to usb mode and adjust C97 for 453.5kHz.

Switch to cw mode and adjust C96 for 454.25kHz.

Switch to fm mode and set the free-running vco frequency of IC3 to approximately 455kHz by adjusting C98. Check that the frequency is similar in the a.m. mode.

(3) Set the level of injection into IC2 to approximately 800mV peak-to-peak by rotating RV1 while in usb mode. Then switch to a.m. and make sure the injection level is similar (but it will now be a square wave, rather than a sine wave).

(4) Now begin testing the audio sections. Connect a 10kΩ resistor to IC5 pin 2 (LK3 is not fitted at this stage) and inject a 1kHz signal into the other end of the 10kΩ resistor. If the generator is not ac coupled, connect a 0.1μF capacitor in series with the 10kΩ resistor. Monitor the audio output of IC8 either by ear or with an oscilloscope. Increase the volume control RV6 and make sure the volume level increases.

By varying the frequency of the generator while each audio filter is selected in turn, some idea of the performance of the filters can be obtained. Figs 3, 4 and 5 show the responses that should be obtained. When checking the narrow filter response, make a note of the frequency at which maximum output is obtained.

When satisfactory performance has been verified, remove the 10kΩ resistor and fit link LK3.

(5) The main 455kHz amplifier (IC1) and the product detector (IC2) can now be tested. Disable the agc action by operating S1, if fitted, or with a wire link across PL3 if S1 is not fitted. Inject a low-level unmodulated 455kHz signal into IC1 pin 4 (via a 0.1μF capacitor if the generator is not ac coupled) and monitor IC2 pin 1. Adjust the core of T6 for maximum signal. The response of T6 is fairly flat and so its core position is not too critical.

Set S2 to usb mode, and monitor IC2 pin 12 (the audio output) or listen to the speaker output. A tone of approximately 1.5kHz should be detected. Switch to lsb mode and the output should be of similar frequency. Now switch to cw mode, the output should be at 750Hz. At this stage C96 can be slightly re-adjusted to set the cw mode audio output to the frequency of the narrow audio filter, as noted in (4).

When these tests have been completed satisfactorily, solder in link LK2.

(6) Now the 455kHz filters and a.m. and fm demodulation can be checked. Inject a low level unmodulated 455kHz signal into the junction of C11/C17/C22 (link LK1 is not fitted). Switch to usb, lsb and cw mode in turn and check that D2 has approximately +6V applied to it at the junction of R33/C26/D2 in these modes. Check that D1 is not conducting while in these modes. Audio outputs as described in (5) should again be obtained.

Now switch to a.m. mode and check that D1 is conducting and D2 is not. If the input frequency is varied on either side of 455kHz, the vco of IC3 should track it, as measured at IC2 pin 10. This shows that the fm signal path, via F1, and probably IC3, are working correctly.

While still in the a.m. mode, if amplitude modulation can be applied to the input signal, recovered audio at the modulating frequency should be obtained.

FM demodulation can be checked if fm can be applied to the input signal

and fm mode is selected. At this stage, operation of the fm squelch circuit can be verified by adjusting the fm squelch control (RV5) while alternately applying and removing the input signal. A position should be found for RV5 where audio output is obtained and LED1 (if fitted) lights when the input signal is applied and a quiet, muted output with LED1 extinguished is obtained with no signal input.

Link LK1 can now be fitted.

(7) Inject a low level unmodulated 10.7MHz signal into the 10.7MHz input (PL1). This input is very sensitive and a resistive potential divider may be necessary to reduce the input level sufficiently. Monitor T3 pin 6 and adjust the core of T3 for maximum signal at 455kHz or maximum audio output. Then adjust the cores of T1 and T2 for maximum 455kHz signal at T3 pin 6. Adjust T1 and T2 cores alternately as adjustments of one may affect the other.

The tests with modulated signals, as described in (5) can now be repeated if desired.

(8) AGC operation can now be checked and the S-meter calibrated. The exact setting of the agc threshold control (RV2) and the S-meter calibration controls (RV3 and RV4) should be completed when the strip is installed in a receiver, but confidence can be obtained at this stage that they are capable of operating satisfactorily.

Enable the agc by operating S1 or removing the link across PL3. Do not fit the S-meter yet. Set S3 to usb mode. Monitor the dc voltage on IC10 pin 2 (connector PL10 is a convenient point). With no signal input, the voltage here should be close to 0V. Now measure the voltage on IC9 pin 7 (or PL11); this should be close to 5V.

Apply a low-level unmodulated 10.7MHz signal to PL1 and slowly increase the level. The voltage on IC9 pin 7 should start to increase in sympathy with the increase in input signal level after a certain threshold level has been reached. The threshold level should be adjustable by varying the setting of RV2. If the audio output is monitored, it should remain almost constant in amplitude after the agc threshold has been reached despite a large increase in the input level. This verifies that the gain of IC1 is being reduced as the input signal level increases.

With the input signal level set at a level beyond the agc threshold, remove the input suddenly while monitoring IC10 pin 2. The voltage should remain at its old level for approximately 1s and then rapidly fall to close to 0V. This is the "hang" facility which IC2 provides, holding the gain of IC1 constant during inter-syllabic and inter-word breaks in transmission and returning to high gain when a break is long enough to indicate a probable end of transmission. Now fit the S-meter. With no input signal, adjust RV3 for zero reading on the meter. RV4 can now be adjusted for full-scale deflection on the meter when a strong signal is being received. This can only be done when the strip has been connected to a front-end. Readings from the S-meter should be used for comparison purposes rather than as an absolute indication of signal strength.

This completes the testing and alignment procedure.

Fault tracing may be simplified by referring to the typical voltages measured on the transistors and ICs of the prototype. These are shown in Table 1.

Alternatives

With such a comprehensive design there are many alternative or sub-equipped versions which can be built to suit individual preferences. Some are described here.

The advantages of 9MHz as a first i.f. may be attractive, and this can be catered for by tuning T1 and T2 to this frequency. They will tune to 9MHz simply by core adjustment, and no modifications to the windings are necessary. X1 will need to be changed to 9.455MHz and T4 tuned to this frequency.

The agc-off facility can be omitted by not connecting S1. Alternatively, a manual i.f. gain control can be incorporated by switching pin 5 of IC1 to the wiper of a potentiometer having potentials of 5 and 7V at each end. If any overloading effects are encountered, the gain of the input amplifier can be reduced by removing C3.

AGC can be applied to the 10.7MHz stage without modifying the pcb by using a PIN diode (such as a BA379) variable attenuator prior to TR1. A description of the use of PIN diode attenuators is given in [12].

For some applications, single-conversion starting at 455kHz is sufficient. In this case, TR1, TR3, T1, T2, T4 and associated components should be omitted. TR2 and T3 can be retained if required to act as a buffer amplifier and to provide 455kHz selectivity. TR2 will require a 100kΩ resistor from its gate to 0V for dc bias, and the 455kHz input coupling through a 470pF capacitor. Alternatively, TR2 can be used as the conversion mixer from rf to i.f. Inject the local oscillator into the source of TR2.

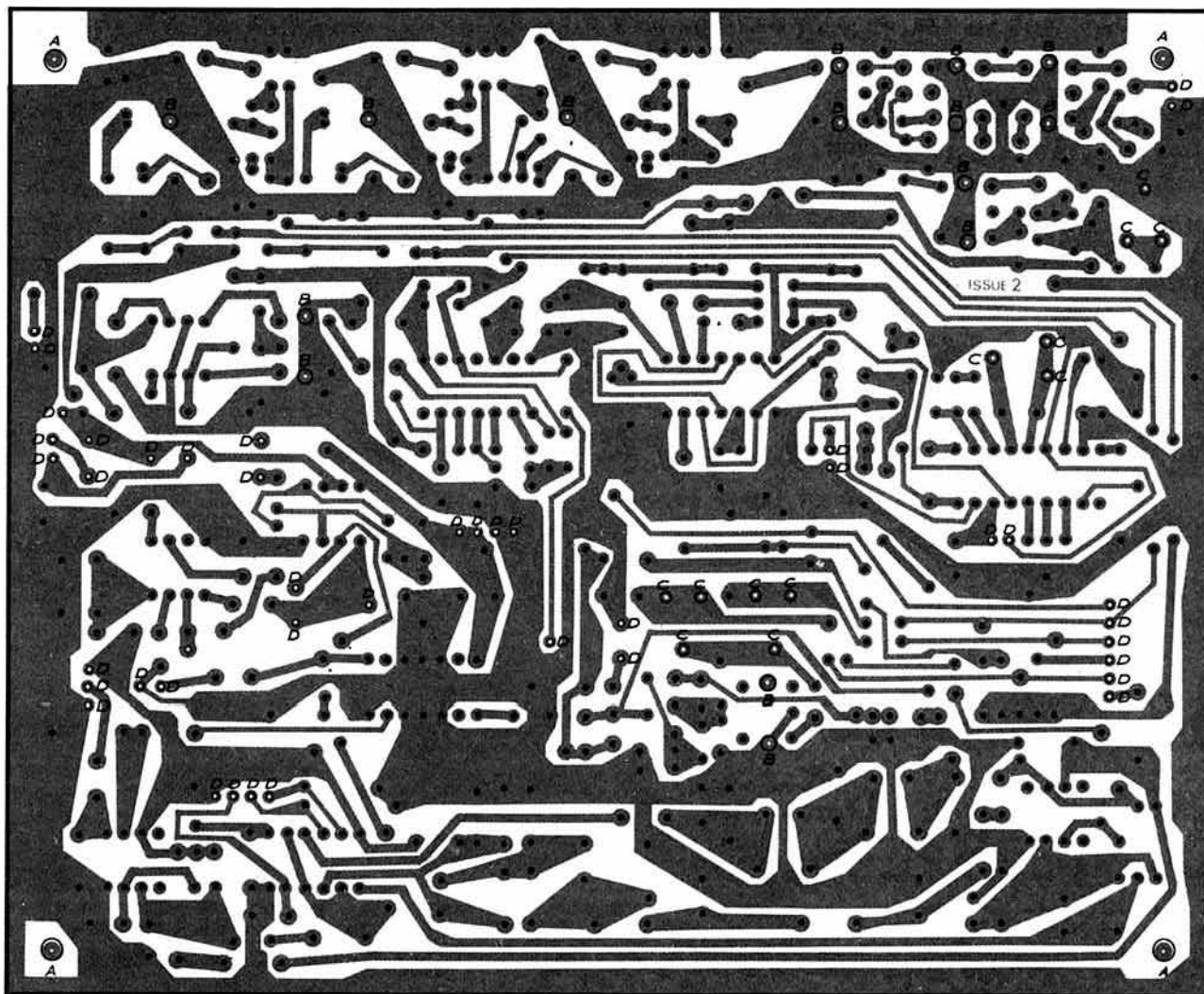


Fig 7. PCB drilling plan. Holes shown • are 0.85mm diameter. Four holes "A" are 3mm diameter. 15 holes "B" are 1.6mm diameter. 12 holes "C" are 1.3mm diameter. 43 holes "D" are 1mm diameter

Components list

R1, 36, 37, 60, 70, 92, 95, 96	100kΩ	R58, 89	470kΩ	C43, 57, 59, 77, 85	10μF tb 16V min
R2, 4, 8, 16, 42, 43, 50	100Ω	R63	2.2kΩ	C46	33pF cp 63V
R3	3.3kΩ	R91	2.7Ω	C47	1.5nF cp 63V
R5	3.9kΩ	All resistors are 0.25W 5% carbon film type			6.8pF cp 63V
R6, 7	15kΩ	RV1	1kΩ	C51	1μF tb 16V min
R9	5.6kΩ	RV2	470kΩ	C62, 66	47nF poly
R10, 18, 26, 65, 68	12kΩ	RV3, 4	10kΩ	C63	4.7nF poly
R11, 19, 27	8.2kΩ	RV5	100kΩ Lin	C64	56nF poly
R12, 15, 20, 23, 29	1.5kΩ	RV6	10kΩ Log	C65	12nF poly
R13, 21, 46, 48, 49, 61, 64, 74, 75, 76, 77, 93	1kΩ	XI	10.245MHz HC18U	C67, 69, 75	0.22μF poly
R14, 22	220Ω	S1	SP, single make toggle	C68	0.18μF poly
R17, 94	470Ω	S2	SP, five-way rotary	C70, 74	0.1μF poly
R24, 25, 32, 33, 34, 35, 51, 66, 67, 69, 72, 73, 80, 81, 82, 83, 84, 85, 86, 87, 88, 97	10kΩ	S3	SP, three-way rotary	C71	10nF poly
R28, 31, 78, 79	2kΩ	M1	200μA S-meter	C72	0.12μF poly
R30, 62	330Ω	C1, 9, 28, 53	470pF cp	C73	27nF poly
R54	680Ω	C2, 4, 6, 8, 14, 15, 27, 32, 34, 36, 37, 38, 39, 40, 44, 45, 48, 50, 54, 55, 56, 58, 60, 61, 76, 78, 79, 82, 84, 86, 91, 92, 93, 94	0.1μF dc (see text) 30V	C80	4.7μF tb 16V min
R38	1.8kΩ	C3, 11, 12, 13, 16, 17, 18, 19, 20, 21, 22, 23, 24, 25, 26	10nF dc 30V	C81, 83, 87	470μF rle 16V min
R39	560Ω	C5, 31, 33	68pF cp 63V	C88, 90	100μF tb 6.3V min
R40	68kΩ	C7	2.2nF cp 63V	C89	47μF tb 6.3V min
R41, 56, 59	4.7kΩ	C10	100pF cp 63V	C95, 96, 97, 98	5-60pF pdt
R44	1.2kΩ	C29	22nF c 63V	dc = disc ceramic	
R45	820Ω	C30	1nF cp 63V	cp = ceramic plate	
R47	51Ω	C35, 41, 42, 52	4.7nF cp 63V	c = ceramic	
R52, 53	2.7kΩ			tb = tantalum bean	
R55, 71, 90	22kΩ			poly = polyester (Siemens type)	
R57	33kΩ			pdt = polypropylene dielectric trimmer	
				rle = radial lead electrolytic	

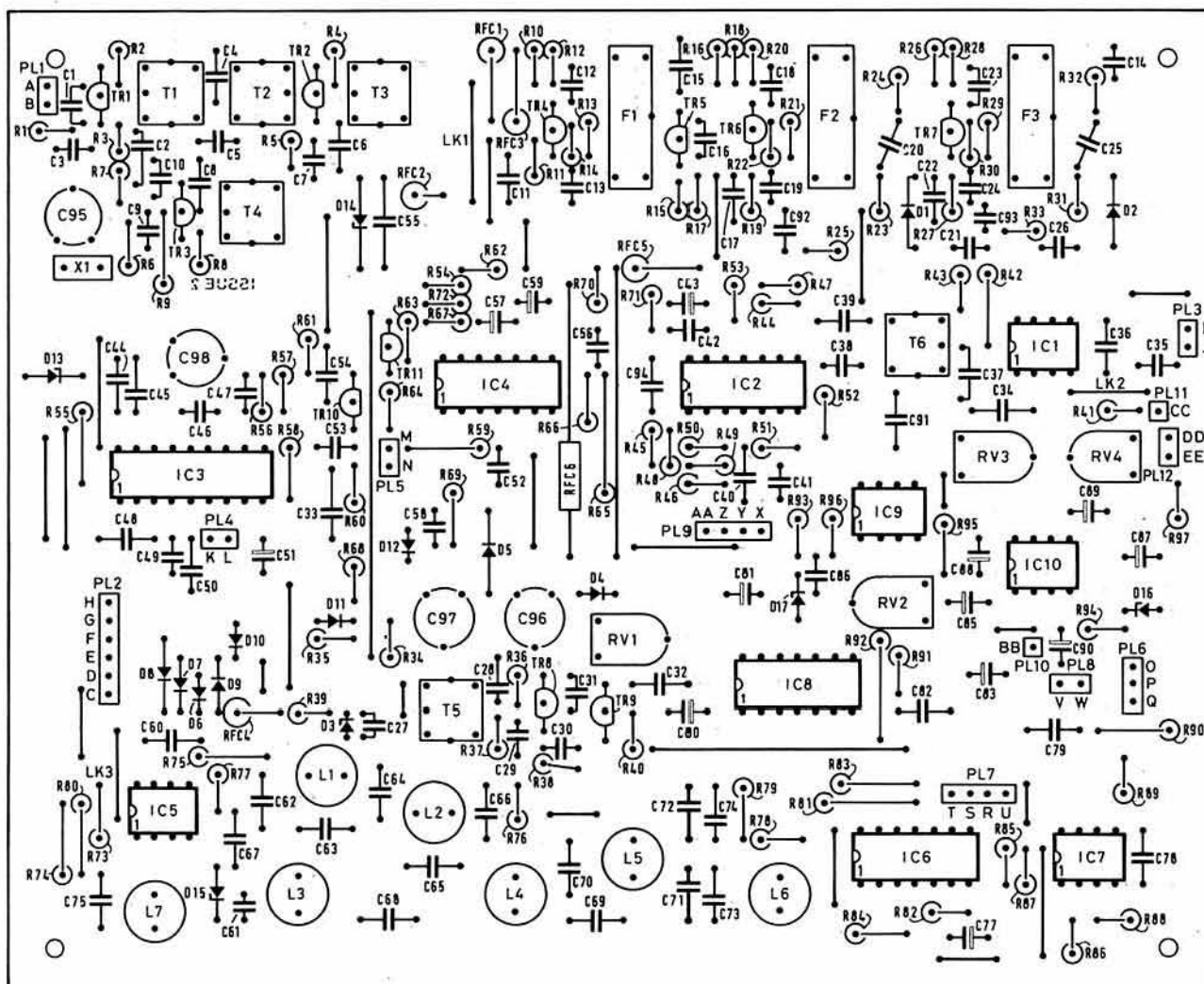


Fig 8. PCB component layout

PL1, 3, 4, 5, 8, 12	Two-way	} 0.1in pitch pcb plug, cable shell
PL2	Six-way	
PL6	Three-way	
PL7, 9	Four-way	
PL10, 11	One-way	
L1, 2	Toko 22mH type 10RB	
L3, 4	Toko 470mH type 10RBH	
L5, 6	Toko 47mH type 10RB	
L7	Toko 220mH type 10RBH	
F1	Murata CFM455F or NTK LFC12	
F2	Murata CFM455G or NTK LFC8	
F3	Murata CFM455J1	
IC1	MC1350 (Motorola)	
IC2	MC1496 (Motorola or National)	
IC3	SL6600 (Plessey)	
IC4, 6	CD4066B (Various Mfrs)	
IC5	TLO71 (Texas)	
IC7	CA3080 (RCA)	
IC8	LM380 (National)	
IC9	LM1458 (Various mfrs)	
IC10	SL1621 (Plessey)	
D1, 2, 4, 5	BA244	
D3, 13, 15	6-2V zener diode	
D6, 7, 8, 9, 10, 11, 12	IN4148	
D14, 16	6-8V zener diode	
D17	9-1V zener diode	
TR1, 2, 5, 10	BF245	
TR3, 4, 6, 7	BF241	
TR8, 9, 11	BC238	
Sockets 5-off, 8-pin dil ic. 4-off, 14-pin dil ic. 1-off, 18-pin dil ic.		
31-off, crimp		

Not all modulation modes need to be catered for. A.M. can be omitted by leaving out TR6, F2 and associated components and removing the a.m. position on S2. If the fm mode is removed, by omitting TR4, TR6, F2, TR10, TR11, IC3, IC4 and associated components, a.m. will have to go as well because of the method used to demodulate a.m. A strap will be required between pins 8 and 9 of the vacated IC4 position to allow the ssb/cw audio into the audio stages. If this is stripped down, ssb/cw-only version is constructed, the wide-bandwidth audio filter may be omitted as well. If ssb only is required, the narrow audio filter along with IC6 and S3 can also be omitted. Strap together pins 3 and 4 of the IC6 position to allow the filtered audio into the volume control, IC7.

A conventional volume control can be used insted of IC7. However, the

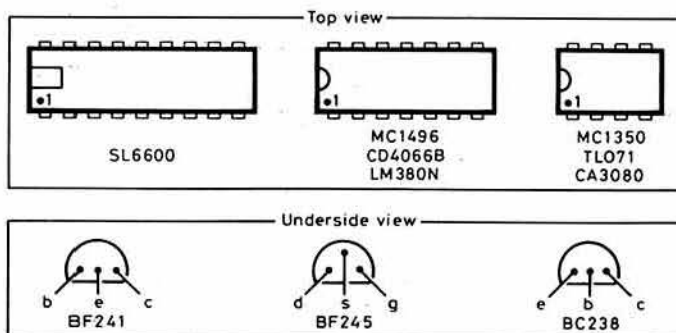


Fig 9. Component pin-outs

Table 1. Voltage check chart

Voltages measured with volume and FM squelch controls fully anti-clockwise. No signal input, no audio output. Supply equal to 12V. All voltages measured with high impedance dvm. Voltages marked * are at floating points and may differ greatly from those measured without malfunction.

TRANSISTORS

TR	Emitter	Base	Collect	Source	Gate	Drain
1	—	—	—	1.2	0	12
2	—	—	—	1.4	0	12
3	4.6	5.2	11.9	—	—	—
4	3.7	4.4	6.4	—	—	—
5	—	—	—	1.0	0	11.7
6	3.8	4.5	6.3	—	—	—
7	4.0	4.6	6.7	—	—	—
8	2.3	3.0	6.3	—	—	—
9	9.2	9.9	12.0	—	—	—
10	—	—	—	0	0	6.9
11	6.0	6.6	6.7	—	—	—

USB, LSB or CW MODE.
COMMUNICATIONS
AUDIO FILTER SELECTED

TRANSISTORS

TR	Emitter	Base	Collect	Source	Gate	Drain
1	—	—	—	1.2	0	12
2	—	—	—	1.4	0	12
3	4.6	5.2	11.9	—	—	—
4	3.7	4.4	6.4	—	—	—
5	—	—	—	1.0	0	11.7
6	3.8	4.5	6.3	—	—	—
7	4.0	4.6	6.7	—	—	—
8	0	0	0	—	—	—
9	9.2	9.9	12.0	—	—	—
10	—	—	—	0	0	6.9
11	0.2	0.3	6.7	—	—	—

FM MODE NARROW
AUDIO FILTER
SELECTED

INTEGRATED CIRCUITS

IC	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18
1	11.3	11.0	0	3.7	5.3	7.3	0	11.3	—	—	—	—	—	—	—	—	—	—
2	4.0	3.3	3.3	4.0	1.2	9.3	0	7.2	0	7.2	0	9.2	0	0	—	—	—	—
3	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
4	0	6.6	0	0	6.6	4.9	0	2.2	2.2	2.4	0	0	0	6.7	—	—	—	—
5	0	6.6	6.6	0	0	6.6	12.0	0	—	—	—	—	—	—	—	—	—	—
6	6.6	6.1	6.1	6.1	12.0	0	0	12.0	12.0	6.1	3.2	0	0	12.0	—	—	—	—
7	0	6.0	6.0	0	0	0	12.0	0	—	—	—	—	—	—	—	—	—	—
8	6.2	0	0	0	0	0	5.8	0	0	0	0	0	12.0	—	—	—	—	—
9	10.6	9.0	9.2	0	4.7	4.7	4.7	12.0	—	—	—	—	—	—	—	—	—	—
10	1.2	0.2	1.0	7.1	1.0	0.5	0	0	—	—	—	—	—	—	—	—	—	—

TRANSISTORS

TR	Emitter	Base	Collect	Source	Gate	Drain
1	—	—	—	1.2	0	12
2	—	—	—	1.4	0	12
3	4.6	5.2	11.9	—	—	—
4	3.7	4.4	6.4	—	—	—
5	—	—	—	1.0	0	11.7
6	3.8	4.5	6.3	—	—	—
7	4.0	6.6	6.7	—	—	—
8	0	0	0	—	—	—
9	9.2	9.9	12.0	—	—	—
10	—	—	—	0	0	6.9
11	0.2	0.3	6.7	—	—	—

A.M. MODE. WIDE AUDIO
FILTER SELECTED

INTEGRATED CIRCUITS

IC	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18
1	11.3	11.0	0	3.7	5.3	3.7	0	11.3	—	—	—	—	—	—	—	—	—	—
2	4.0	3.3	3.3	4.0	1.2	9.3	0	7.2	0	7.2	0	9.2	0	0	—	—	—	—
3	0.8	0.8	1.2	2.4	1.7	6.3	5.3	4.7	6.2	6.2	6.0	6.1	4.9	4.9	6.4	3.3	0	0.7
4	0	0.3	4.7	2.0	0.3	5.2	0	2.2	2.2	6.0	2.0	0	6.3	6.7	—	—	—	—
5	0	6.6	6.6	0	0	6.6	12.0	0	—	—	—	—	—	—	—	—	—	—
6	6.6	3.2	6.0	3.2	0	0	0	12.0	12.0	3.2	3.2	12.0	0	12.0	—	—	—	—
7	0	6.0	6.0	0	0	0	12.0	0	—	—	—	—	—	—	—	—	—	—
8	6.2	0	0	0	0	0	5.8	0	0	0	0	0	12.0	—	—	—	—	—
9	10.6	9.0	9.2	0	4.7	4.7	4.7	12.0	—	—	—	—	—	—	—	—	—	—
10	1.2	0.2	1.0	7.1	1.0	0.5	0	0	—	—	—	—	—	—	—	—	—	—

INTEGRATED CIRCUITS

IC	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18
1	11.3	11.0	0	3.7	5.3	3.7	0	11.3	—	—	—	—	—	—	—	—	—	—
2	4.0	3.3	3.3	4.0	1.2	9.3	0	7.2	0	7.2	0	9.2	0	0	—	—	—	—
3	0.8	0.8	1.2	2.4	1.7	6.3	5.3	4.7	6.2	6.2	6.0	6.1	4.9	4.9	6.4	3.3	0	0.7
4	0	0.3	4.7	2.0	0.3	0	0	2.2	2.2	2.7	2.0	5.4	6.3	6.7	—	—	—	—
5	0	6.6	6.6	0	0	6.6	12.0	0	—	—	—	—	—	—	—	—	—	—
6	6.6	6.6	6.1	6.6	0	0	0	12.0	12.0	6.6	3.2	0	12.0	12.0	—	—	—	—
7	0	6.0	6.0	0	0	0	12.0	0	—	—	—	—	—	—	—	—	—	—
8	6.2	0	0	0	0	0	5.8	0	0	0	0	0	12.0	—	—	—	—	—
9	10.6	9.0	9.2	0	4.7	4.7	4.7	12.0	—	—	—	—	—	—	—	—	—	—
10	1.2	0.2	1.0	7.1	1.0	0.5	0	0	—	—	—	—	—	—	—	—	—	—

extra cost of IC7 is so minimal as to make this alternative not really worth considering.

The audio stages of this strip, from IC5 to the loudspeaker driver, could be used without the i.f. stages. The communications filter in particular is good enough (and cheap enough) to be incorporated into many other designs.

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